# TONE CONTROL, VOLUME EXPANSION AND NOISE LIMITING CIRCUITS

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#### STUDY SCHEDULE NO. 24

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions specified for that step. Study each other step in this same way.

	۱.	Satisfying the Human Ear
□ 2	2.	Simple Tone Controls; Special Bass and Treble Tone Controls Pages 5-9 Four-Position Tone Control; What Is Best Tone Control Setting; Continuously Variable Tone Control; How Tone Controls Reduce Noise; How Tone Controls Affect A.F. Response Curves; Parallel Resonant Bass Control Circuit; Audio Input-Shunting Bass Control Circuit; Treble Controls. Answer Lesson Questions 3 and 4.
<u> </u>	3.	Automatic Tone Controls; Unique Tone Controls. Pages 9-15 Condenser-Type A.B.C. Circuit; Series-Resonant A.B.C. Circuit; Separate Bass and Treble Controls; Motorola Acoustinator; G-E Tone Monitor. Answer Lesson Questions 5 and 6.
□ 4	ŀ.	Volume Expansion Circuits
<u> </u>	5.	The Noise Problem; Noise-Reducing Tone Controls; Noise Impulse-Silencing Circuits
☐ 6	5.	Inter-Carrier Noise Suppression Circuits Pages 24-28 Biased Demodulator Circuit; A.F. Amplifier Blocking Circuit; I.F. and R.F. Amplifier Blocking Circuit. Answer Lesson Questions 9 and 10.
□ 7	7.	Mail Your Answers for this Lesson to N.R.I. for Grading.
<u> </u>	3.	Start Studying the Next Lesson.

### Tone Control, Volume Expansion and Noise Limiting Circuits

#### Satisfying the Human Ear

THE manufacturers and operators of sound broadcast transmitters have one objective in common with the manufacturers of radio receivers—to make the receiver owner feel that he is listening to the actual studio performance rather than to a loudspeaker. This is by no means a simple task, and as yet has not been completely achieved even in the most expensive radio receivers. Radio engineers are doing everything within their power to improve the faithfulness of reproduction, and each year brings them another step closer towards their goal of perfection.

In this text-book we shall consider the various factors which make it difficult to achieve complete perfection, and study the methods used to satisfy the human ear under different conditions.

Distortion. One important requirement for faithful reproduction is freedom from all types of distortion. All of the sound frequencies which are present in the original performance must be present in the reproduced program and must seem to have the same loudness relation to each other as in the original performance; furthermore, there must be no frequencies in the reproduced program which did not originally exist (that is, there must be negligible amplitude distortion and negligible noise originating in the transmitting and receiving apparatus).

Noise. Broadcast transmitters can be, and in most cases are designed to radiate negligible noise. Radio receivers can likewise be so designed that noise originating in them is negligible. It is noise which originates outside of the transmitting and receiving apparatus which requires special attention. Man-made interference noise is the chief offender in this respect; for this reason every serviceman should be familiar with the use of noise-reducing antennas for receivers and the application of noise filters to trouble-making electrical devices, all of which are taken up elsewhere in this Course.

Atmospheric noises, which enter the receiver along with the carrier of the desired station, cannot be entirely eliminated, but their annoying effects. can be reduced considerably. You will learn that there are three practical types of circuits used for this purpose: 1, Tone Control circuits, which cut out high-frequency sound components, eliminating predominating noise signals at a sacrifice of fidelity; noise impulse-silencing circuits, which temporarily cut out all signals for the duration of sharp noise pulses which are stronger than the program signals; 3, inter-carrier noise suppression circuits, which cut out all signals, including noise, whenever the station signal is so weak that it is drowned out by noise signals. (Increasing the amount of power radiated by a transmitter, in order that the desired signal can better override noise signals, is a solution, but transmitting power is definitely limited by law in order to prevent interference between stations.) Typical examples of the ingenious circuits which have been developed to accomplish these results will be studied in this lesson.

Peculiarities of the Human Ear. The characteristics of the human ear have

a lot to do with radio receiver design. Possibly you have noticed that orchestra music which sounds fine at a moderate distance away seems high-pitched (lacking in low frequencies) when you walk a considerable distance away.

The same phenomenon occurs during the fading of a radio program; this can most readily be noticed when a foreign short-wave program is tuned in. At a normal listening level of loudness the music will sound fine, but as the signal fades out, the music seems to become high-pitched or squeaky; as signal strength comes up to normal again, the music gradually becomes more mellow and pleasing.

The reason for this puzzling effect is simply that our ears hear low frequency or bass notes better at medium and high loudness levels than at low loudness levels. When the volume of a radio program drops because of fading, the bass notes seem to be cut far more than the higher frequencies.

Since radio receivers in homes are ordinarily operated at a loudness level considerably lower than that of the original performance, we have the queer situation that even a perfect radio receiver (one having a flat response over the entire audio range) would sound unsatisfactory whenever its output volume differed from that of the original program.

Characteristic Curves for the Average Human Ear. The peculiar hearing characteristics of the human ear have been carefully investigated. The results of tests on thousands of persons have been combined in the graph in Fig. 1, which tells how the average ear responds to various frequencies at different loudness levels.

A few words of explanation as to how data for this graph was secured will help you to read its story. First of all, a 1,000-cycle sound was varied in loudness, while various persons listened, until an average loudness level which could just barely be heard was determined. This was called the 0 (zero) db loudness level. The actual intensity of this 1,000-cycle sound was then determined with a sound-measuring instrument, and the meter scale on the instrument was made to indicate 0 db intensity level for this condition. Now the same test was repeated for other frequencies, and the intensity level in db at which the sound could just barely be heard was measured with the instrument. Thus, at 500 cycles the average meter reading was +6 db; at 100 cycles it read 38 db, and at 30 cycles the reading was +63 db, etc. Each value was plotted on the graph, and a smooth curve was drawn through all the points to give the 0 db loudness level curve, also known as the threshold of hearing curve. This curve shows clearly that intensity levels considerably greater than 0 db are required below 1,000 cycles in order for sound to be heard, and also shows that the human ear hears best at about 4.-000 cycles per second.

The intensity level at 1,000 cycles was then raised 10 db to give a loudness level of 10 db at this frequency, and the intensity level at each other frequency which seemed to give this same loudness was determined. This gave data for the 10 db loudness level curve. The same procedure was repeated at 10 db intervals to secure data for the remaining curves in Fig. 1.

It was found that when the loudness level was made higher than 120 db, the sounds were actually felt by the persons taking the test, the sensation being that of pain in the ears in some instances; the 120 db loudness level curve is therefore known as the threshold of feeling.

How to Use the Ear Characteristics Graph. Suppose we want to determine the intensity level required for a certain loudness, say 20 db, at a given frequency such as 200 cycles. We simply locate this frequency on the horizontal scale (point A), trace directly upward from it to the point where that frequency line intersects the 20 db loudness curve (point B), and then trace horizontally to the vertical reference scale at the left, where we read +40 db (point C) as the intensity level required in db for that loudness. In other words, for a 20 db loudness level

microphone which picks up all sound frequencies equally well.

We can draw these general conclusions from the curves in Fig. 1: At low loudness levels (from 20 to 40 db), such as those ordinarily produced by a radio receiver loud-speaker in the home, the human ear is most sensitive to sounds in the middle frequency range from 500 cycles to 5,000 cycles, and has difficulty in hearing sounds below and above this frequency range.

As we raise the *loudness level* of sound, such as by turning up the volume control on the receiver, the sensi-

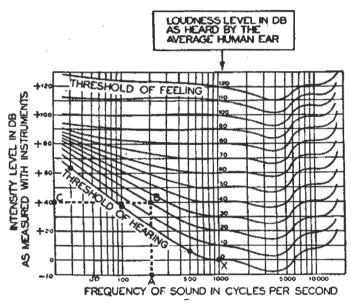


Fig. 1. This standard reference graph gives the average hearing characteristics (frequency response) of the human ear at various loudness levels; in other words, it tells what intensity level will be required for a given loudness level at a given frequency.

our ear requires 20 db more sound power at 200 cycles than at 1,000 cycles. At a 90 db loudness level, however, the curve is practically horizontal between these two frequencies, indicating that our ear can hear both frequencies equally well when they are this loud.

Bear in mind that loudness level is a human response to sound, and not something which can be measured with instruments. Intensity level is the actual measured level of sound as measured by instruments connected to a tivity of the ear to low frequencies improves rapidly, so that at a loudness level of 100 db we can readily hear all sound frequencies below 10,000 cycles equally well. The lower the volume control setting on a radio receiver having an ordinary volume control, then, the fewer low or bass notes there will be insofar as our ears are concerned; to a lesser extent this statement also applies to the extreme high frequencies.

To get true high-fidelity reproduction at any desired loudness level, then, it is necessary to compensate for the shortcomings of the human ear by boosting the response of a receiver at the low and high frequencies the correct amount for each loudness level. Boosting of the lows (raising of the bass notes) is far more important than raising the high-frequency or treble sounds.

Tone Controls. Many receivers are provided with separate bass and treble tone controls, which allow the listener to adjust the response of the receiver to secure faithful reproduction or to suit his particular taste at any desired loudness level. Oftentimes, however, the radio engineer makes these tone controls entirely automatic in operation, so that as volume is reduced, the low and the high frequency notes are automatically strengthened with respect to the middle frequencies. The amount of tone compensation provided at each loudness level is governed by the average reaction of the human ear to sound; receiver design engineers continually refer to the standard curves in Fig. 1 for this purpose. Automatic tone control is particularly appreciated by those listeners who do not have the ability to judge the correct adjustments of the bass and treble controls, or who do not want to take the time required to make these adjustments. Both manual and automatic tone controls will be taken up in this lesson.

Volume Compression. Of course, no radio program is broadcast at a constant volume level from instant to instant. For example, a symphony orchestra may have a variation in intensity of as much as 100 db from the weakest to the strongest passages. If the transmitter were adjusted to handle the loudest passages without overmodulation, then the weakest passages would be completely drowned out by transmitter circuit noise, and

the average modulation percentage will be too low for economical operation; on the other hand, if the transmitter is set to keep the lowest passages well above the transmitter circuit noise level, then severe overmodulation and distortion would occur on loud passages. A maximum variation of 40 db is about the most a transmitter can handle, although a range of 30 db seems to be preferred in most transmitters. If the signal level is allowed to rise too high, over-modulation and distortion occur; oftentimes this also results in overloading of the transmitter and opening of circuit breakers, throwing the station temporarily off the air. It is the duty of the studio operator who monitors a broadcast to reduce the audio gain of the transmitter for loud passages and boost the gain for weak passages, thereby compressing the sound into the allowable volume range. Automatic volume limiting systems are also employed for this purpose; these compress just before over-modulation occurs, by cutting down gain.

Volume Expansion. The compression of the original sound into a limited volume range will reduce the quality of the transmitted program. In a number of radio receivers special volumeexpanding circuits are used to counteract this compression at high levels and increase the range of volume somewhat. Full expansion is ordinarily not possible because of the inability of the amplifier in the receiver to furnish the power required for the loudest sounds produced in the studio and because only a part (or none) of the compression occurring at the transmitter is automatic, but there is an appreciable improvement in the quality of reproduction when a volume-expanding circuit is incorporated in a receiver.

Sounds are compressed in much the

same way when phonograph records are made, and many of the better phonograph amplifiers have volume-expanding circuits. Because the high level compression usually is automatic in recording studios, the expansion can be made to correspond almost exactly, and very good fidelity can therefore be secured. Several different automatic volume expansion circuits will be studied.

Directional Effects. When we listen to an actual orchestra, our ears are able to determine from which direction each particular sound comes; thus we may note that the drums are to the right of the conductor, while the bass horns are at his left. When all these sounds are reproduced by a single loud-speaker at the receiving point, however, we lose this effect completely, and all of the sounds appear to come from the loudspeaker. With our present system of broadcasting, it is impractical to correct this situation, although it has been done already on an experimental basis.

In one test broadcast, two microphones were located a definite distance apart in the studio, and the sounds picked up by each were independently amplified and broadcast over two separate transmitters. At the receiving location were two separate receivers and loudspeakers, with the loudspeakers located the same distance apart as the microphones in the studio. As a result, the actual directional characteristics of the orchestra were duplicated.

Curiously enough, the lack of directional qualities in a radio program in no way destroys the entertainment value of the program for the average listener. We have become accustomed to radio transmission as it is, and consequently the tremendous expense involved in complete duplicate transmitting and receiving systems for each

broadcast is unwarranted at the present time.

#### Simple Tone Controls

Any device which, when introduced into a radio receiving circuit, serves to reduce or remove the higher-frequency sound signals will have the apparent effect of boosting the low and medium frequencies. A condenser is one simple device for doing this, and is therefore widely used in simple tone controls. It is customary to make the insertion of one or more condensers in the circuit for tone control purposes optional, so the listener can select the most satisfactory tone control position for each loudness level.

When a tone control condenser is connected between the grid and the cathode or chassis of an audio amplifier stage or between the plate and the cathode or chassis, the high reactance of this condenser at low frequencies results in negligible by-passing of the signal, but at medium and high frequencies the reactance of the condenser is so low that there is appreciable signal by-passing. A simple condenser tone control gives an apparent boost in bass or low-frequency signals because the signal which it feeds to the next stage has a higher proportion of bass notes (with respect to medium and treble notes) than the signal fed into the tone control.

Simple Four-Position Tone Control. One simple and widely used tone control arrangement is that shown in Fig. 2A, where selector switch SW serves as the tone control. When this switch is set at point 1, no shunting or by-passing of the signal occurs, and hence all frequencies in the sound signal are fed to  $C_K$  without attenuation. As the switch is set to points 2, 3 and 4, increasingly greater capacities are placed between the plate and ground, and in-

creasing amounts of the middle and high frequencies are shunted to ground without passing through the load resistor  $R_{\rm L}$  for transfer to the following stage.

In inexpensive receivers there are generally only two positions of the tone control, and only one condenser; in the first position of the tone control switch, which might be labeled "BRIGHT," no condenser is in the circuit and all frequencies are passed uniformly. In the second position of the switch, often labeled "MELLOW," the condenser is in the circuit and the medium and higher frequencies are cut

that they will not sound any louder to the human ear than the low and medium frequency signals. It must be remembered that the curves in Fig. 1 were secured by averaging the hearing characteristics of a great many persons. Individuals may therefore deviate considerably from these curves.

Continuously Variable Tone Control. In the circuit of Fig. 2A, the tone control must be adjusted in steps, with the result that proper correction of tone will not be possible at certain loudness levels in between these steps. For this reason many people prefer a continuously variable tone control like

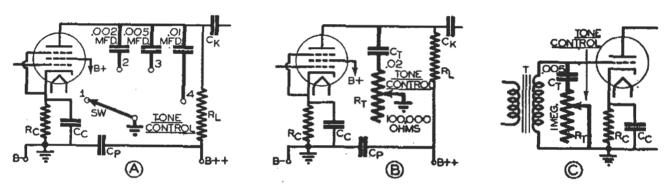


Fig. 2. Simple manual tone control circuits which utilize the signal by-passing action of a condenser,

down, giving a bass-boosting effect. Receivers with three-position tone controls, labeled "BRILLIANT," "BRIGHT" and "MELLOW," would use two condensers.

What Is the Best Tone Control Setting? It is incorrect to say that highest fidelity is always obtained at one particular position of the tone control. When the receiver is turned to maximum volume, giving a high level of loudness, that position in which there is no condenser in the circuit to by-pass the higher frequencies will give the best fidelity if the receiver normally has a flat response. On the other hand, when the receiver volume is turned down, it may be necessary to place the highest-capacity tone control condenser in the circuit in order to reduce the highest frequency signals enough so

that shown in Fig. 2B. Here a variable resistor is placed in series with a single tone control condenser. When this resistor is set at zero resistance, only the condenser is connected between the plate of the tube and ground, and we have maximum reduction of the high frequencies (the low and medium frequencies will then come through considerably stronger than the high frequencies). Increasing the resistance increases the net impedance of this tone control path to ground, thus reducing the by-passing effects on the high frequencies. When the maximum tone control resistance of 100,000 ohms is in the circuit, practically no signal frequencies take the path through the tone control circuit and there is no cutting of high frequencies.

A similar continuously variable tone

control arrangement is shown in Fig. **2C** as it would be connected between the grid and chassis of an audio amplified stage. The action of this circuit is identical to that in Fig. 2B; a lower value of capacity and a higher value of resistance are required for the grid connection, as you can see by comparing the values specified in Fig. 2C with those in Fig. 2B. It is important to realize that an appreciable amount of attenuation or cutting of high frequencies can be achieved only if the impedance of the tone control circuit is considerably lower than the impedance of the load which it shunts. Since the grid-cathode path of the tube in Fig. 2C is a high-impedance load for the secondary winding,  $C_{\rm T}$  and  $R_{\rm T}$  have high ohmic values.

How Tone Controls Reduce Noise. Simple tone controls are widely used in all-wave receivers, and are especially valuable when listening to short-wave programs coming from far distant transmitters. The signals from these distant stations are generally accompanied by a great deal of noise; this is because they are received at such low levels that receiver gain must be increased considerably, and noise then becomes more noticeable. It is a known fact that in these noise signals the higher sound frequencies predominate; when the tone control is set to cut down high frequencies, the received signal becomes considerably less noisy. You can easily verify this for yourself by trying the tone control on an allwave receiver while tuned to a European station; note how the noise is reduced when the tone control is set for MELLOW or BASS reception.

Tone Controls May Be Connected Anywhere in the A.F. Amplifier. There appears to be no one preferred position for the tone control in an audio amplifier. Sometimes this control is

connected in the output circuit of a diode detector, right at the input of the audio amplifier, as shown in Fig. 3. Potentiometer  $R_3$  serves as the volume control, with the setting of its movable arm determining the amount of A.F. voltage which is fed through blocking condenser  $C_4$  to the grid of the first audio stage. Tone control potentiometer  $R_2$  is shunted across  $R_3$ , and the movable contact of R<sub>2</sub> is grounded through condenser  $C_8$ . Moving the tone control from position 1 to position 2 gradually increases the shunting effect of  $C_3$ , and therefore increases the cutting or by-passing of high-frequency signals.

A similar tone control arrangement could be used in Fig. 2B by replacing

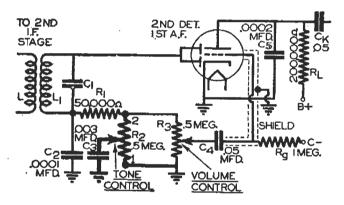


FIG. 3. Simple manual tone control circuit acting on the load of a diode second detector; the parts values are those used in the Silvertone Model 1825A superheterodyne receiver.

resistor  $R_{\rm L}$  with a potentiometer and connecting the movable tap of the potentiometer to the chassis through a .02 mfd. condenser;  $R_{\rm T}$  and  $C_{\rm T}$  would then be removed. Likewise in Fig. 2C the secondary of the audio transformer could be shunted with a .5 to 1 megohm potentiometer, with its movable contact grounded through a condenser, in place of the  $R_{\rm T}$ - $C_{\rm T}$  tone control arrangement shown.

How Tone Controls Affect A.F. Response Curves. The effect of a typical four-position tone control upon the over-all audio frequency response of a

radio receiver is clearly shown by the graph in Fig. 4. Curve 1 is for the tone control setting where there is no shunt capacity to cut down the higher frequencies; this curve therefore represents the normal frequency response of the receiver, indicating the relative output intensity level at each sound frequency. Curves 2, 3 and 4 show how this response is changed as increasingly greater shunt capacity is inserted in the circuit. These curves show clearly how a tone control cuts down the medium and high frequencies in the audio range without affecting the low or bass notes. With variable tone controls like those in Figs. 2B, 2C

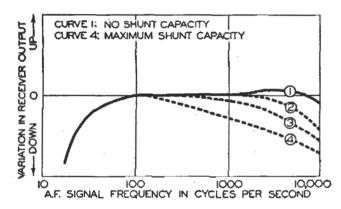


FIG. 4. This graph shows the effect of a four-position tone control upon the over-all frequency response of a typical radio receiver.

and 3, we could have any number of different response curves between 1 and 4, depending upon the control setting.

If the volume level of a receiver is raised to a high value when the tone control setting corresponds to curve 4 in Fig. 4, bass notes will be amplified excessively for the requirements of the human ear, and the over-correction will be readily noticed by the listener as a boomy effect. Some people actually prefer a very strong bass response and will purposely set the tone control for this result, even though they thus destroy the fidelity or quality of reception.

### Special Bass and Treble Tone Controls

In the more expensive radio receivers you will often find two separate tone controls, one for changing the low frequency or bass response and the other for changing the high frequency or treble response. Let us first consider bass controls.

Parallel Resonant Bass Control Circuit. Resonant circuits are widely used to control the bass response of a receiver; a typical example is that in Fig. 5A, where condenser C and iron-core inductance L form a parallel resonant circuit which has a resonant frequency of about 50 cycles (C will ordinarily be about .1 mfd., and L about 100 henrys). At resonance this circuit acts as a high resistance which, being in series with resistor R (about 10,000 ohms), will result in a high plate load impedance. The amplification of the stage is therefore increased in the low frequency or bass range around 50 cvcles.

This parallel resonant circuit acts like a reactance of low ohmic value at frequencies higher than 50 cycles, under which condition resistor R governs the plate load impedance. The greater the resistance of the resonant circuit at resonance, the greater will be the boost in low frequency response; this resonant resistance can be varied over a wide range by adjusting the setting of potentiometer  $R_{\rm B}$ . When all of  $R_{\rm B}$ is in the circuit, the resonant resistance is quite low and there is little or no boosting of bass response. Maximum bass boosting is secured when all of  $R_{\rm B}$  is shorted out.

Audio Input-Shunting Bass Control Circuit. Another widely used bass tone control circuit is shown in Fig. 5B. Inductance L, potentiometer R and condenser C form the bass tone control

circuit, which in this case is not a resonant circuit.

We can neglect the effect of  $C_{K}$ , as its reactance will be small in comparison to the reactance of L at any frequency or in comparison to the resistance of R. Assume for the moment that C is omitted: clearly we now have a voltage divider consisting of R and L, with the reactance between point P and ground determining the voltage output. Since the reactance of L at low frequencies is quite small in comparison to the resistance of R, the bass voltage will be developed essentially across R (between points 1 and 2). As potentiometer contact P is moved from point 2 to 1, less and less of this bass voltage is fed to the grid of the next tube. Since the audio system is intentionally made to have a high bass response, position 2 of the movable tap is for high bass response and position 1 is for low bass response.

Although moving contact P from point 2 to point 1 decreases the amount of bass forwarded to the following tube, it also decreases to some extent the medium and treble frequencies. Condenser C in Fig. 5B is used to bypass medium and high frequency signal components around the upper section of R, and consequently the medium and treble notes are passed by the tone control circuit without appreciable attenuation regardless of the setting of P. This bass control is definitely not a volume control when condenser C is used; if C is omitted, however, volume control action will be present along with bass control.

Treble Controls. The resonant tone control circuit in Fig. 5A can also be made to serve as a treble control if the values of L and C are chosen to provide resonance at a high audio frequency value. With this arrangement, varying the setting of potentiometer

R<sub>B</sub> will vary the high frequency response from its normal value to a value considerably above normal.

Use of a Conventional Tone Control as Treble Control. A more widely used procedure for securing treble control is that wherein the audio amplifier is designed to have a sharp rise in response at the higher frequencies, and one of the conventional tone control circuits shown in Figs. 2A, 2B, 2C and 3 is used to cut down this high frequency or treble response as desired.

#### **Automatic Tone Controls**

The purpose of a tone control, as

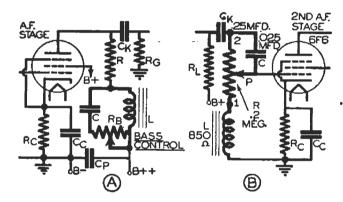


Fig. 5. Typical bass control circuits; that at A can also be made to serve as a treble control, by making L-C resonate at a high audio frequency. That at B is used in a General Electric all-wave receiver.

you already know, is to produce the effect of boosting the extreme low and extreme high-frequency response of the receiver to compensate for the inability of the human ear to hear these frequencies as well as the medium frequencies at low levels of loudness. The same results can of course be accomplished by reducing the level of the middle frequencies more than that of the extreme frequencies when reducing the volume. It is possible to secure this action automatically as the volume control is adjusted by the listener.

Engineers differ as to the extent to which automatic tone control should

correct for the shortcomings of the human ear at low levels of loudness. Some feel that only the bass should be reinforced (automatic bass compensation), while others prefer to reinforce both the treble and the bass (automatic tone compensation) to secure more nearly perfect compensation for the drop in sensitivity of the human ear under the conditions in question.

Automatic bass compensation (often abbreviated as A.B.C.) could be secured with a tone control of the type shown in Fig. 3 if we mounted this tone control potentiometer on the same shaft as the volume control, so that when volume was reduced, the medium and high frequencies would be cut down proportionately in order to satisfy the human ear. While this could be made to work perfectly satisfactorily, there is a much simpler way of accomplishing the same results.

Condenser-Type A.B.C. Circuit. A practical automatic bass compensation circuit is shown in Fig. 6A; although quite simple, it is entirely effective. The two terminals of this circuit, x and y, are connected either across the diode load resistor or across the output of some audio stage in the receiver. Volume control potentiometer  $R_1$  is tapped at point 2, which need not necessarily be its mid-point. Condenser C in series with resistor R between points 2 and 3 serves to attenuate (cut down) the middle and high frequencies at low-volume settings of the control, thus providing automatic bass compensation. Whenever you encounter a circuit like this, where the volume control has a fixed tap which is connected to ground through a condenser and a resistor, you can immediately identify it as an automatic bass compensation circuit.

Consider the circuit in Fig. 6A from the viewpoint of a voltage divider. If C and R were not present, the voltage E at any frequency would depend solely upon the position of contact arm P. With C and R connected as shown, and with the reactance of C decreasing with increases in frequency, the impedance between points 2 and 3 decreases as the frequency increases. The lower this impedance becomes, the more A.F. voltage there will be across section 1-2 of the voltage divider; in fact, at very high audio frequencies almost all of the available signal voltage is developed across points 1 and 2.

When P is at point 1, all of the A.F. voltage developed across the voltage divider is fed to the grid of the next stage regardless of its frequency. If P is placed at point 2, only the voltage developed between points 2 and 3 is fed to the grid of the next stage; the reactance between these points decreases at the higher frequencies and consequently there are fewer high-frequency components than normal in the output voltage E when P is at 2. We thus have the bass-boosting effect required for automatic bass compensation. At any position of P between points 1 and 2, the output voltage E depends upon the ratio of the impedance between P and 3 to the total impedance between points 1 and 3, and the amount of automatic bass compensation increases from zero (when P is at 1) to its maximum value when P is at 2. When P is moved below 2 (towards 3), the amount of bass compensation remains the same as for point 2 and all frequency components are reduced uniformly as in normal volume control action.

Occasionally you may find an additional condenser and resistor con-

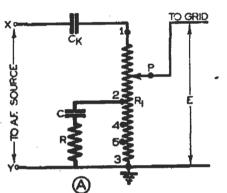
nected in series between point 4 and the chassis, and another condenser and resistor between point  $\delta$  and the chassis; this is done to provide increasingly greater bass compensation as contact arm P is moved from 2 to 3.

Series-Resonant A.B.C. Circuit. Even greater attenuation or cutting of the high and middle frequencies for automatic bass compensation purposes is possible with a series resonant circuit arrangement as shown in Fig. 6B. Coil L and condenser C in series are adjusted to be broadly resonant to the middle and high frequencies; as a result, the effects of their reactances cancel at these frequencies, leaving only resistance R and the coil resistance in shunt with section 2-3 of

sistance to give the desired broad tuning over the middle and high frequency range is used. If the L-C circuit is broadly resonant only for the medium frequencies, only these medium frequencies will cut down at low volume control settings, and automatic tone control (bass and treble compensation) is secured.

#### Unique Tone Control Circuits

Separate Bass and Treble Channels. Some manufacturers of radio receivers provide one path in the audio amplifier for low-frequency signals and another entirely separate path for high-frequency signals; this permits amplification of these signals independently



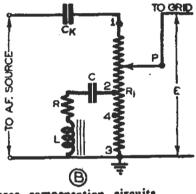


Fig. 6. Two practical automatic base compensation circuits.

potentiometer  $R_1$ ; resistance R prevents complete cutting of signals at the resonant frequency. At low frequencies, however, the reactance of C is so high that the shunting effect of the series resonant circuit can be neglected. As contact arm P is moved from point 1 to point 2, lowering the volume, the amount of cutting of medium and high frequencies gradually increases to a maximum value when Pis at 2. Further reductions in volume (by moving P below point 2) give no further increase in bass compensation unless an additional series resonant circuit or condenser-resistor circuit is connected between point 4 and ground. Very often resistor R is omitted and a coil which has a sufficiently high reand gives a definite control over their relation to each other.

A simple circuit of this nature is shown in Fig. 7. The audio signal output of the first A.F. amplifier tube  $(VT_1)$  divides at point 1; the middle and high frequency signals in the audio range take the path through the .001 mfd. condenser  $C_3$  to audio amplifier tube VTs, but the reactance of this condenser is so high at low frequencies that very few low frequency signals get through. The other path for signals from point 1 is through .5 mfd. condenser C<sub>2</sub> and through 100,000-ohm resistor R to audio amplifier tube  $VT_2$ . Both low and high frequencies can pass through this condenser and resistor, but the high frequencies take

the .05 mfd. shunt path  $(C_4)$  around the .5 megohm potentiometer  $(R_3)$  to ground and consequently do not act upon the grid of  $VT_2$ . Only the low frequency components and a portion of the medium frequencies are amplified by  $VT_2$ . Resistor R prevents  $C_2$ -R- $C_4$  from being a complete short for high frequencies, for that would leave none for the treble tube  $VT_a$ . The potentiometers in the grid circuits of  $VT_2$  and  $VT_3$  can be adjusted to provide any desired relationship between the bass, medium and treble frequencies. The outputs of the bass and treble amplifier tubes are combined at point 2

accentuate the bass and treble frequencies, thus giving high-fidelity reproduction of music at normal low sound levels.

The circuit diagram of a typical Motorola tone control system is shown in Fig. 8. The tone control switch is shown at position V (VOICE). Under this condition the A.F. signal is developed across load resistor  $R_1$ , all R.F. components accompanying this signal being by-passed to ground by the .0002 mfd. shunting condenser  $C_2$ . This audio signal voltage is then applied to .5 megohm resistor  $R_2$  through blocking condenser  $C_3$  and by-pass condenser condenser  $C_3$  and by-pass condenser  $C_3$ .

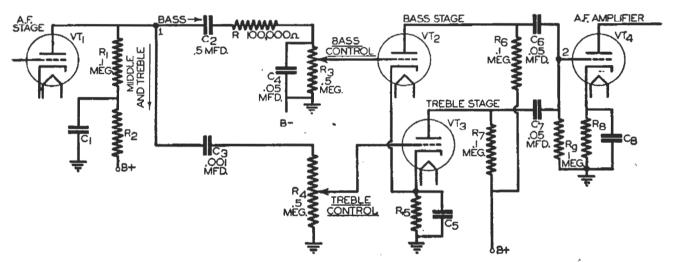


Fig. 7. Special tone control circuit which uses separate channels for amplification of bass and treble frequencies, with independent controls for each channel.

and fed from there to the grid of audio amplifier tube  $VT_4$ .

The Motorola Acoustinator. A rather unique bass and treble tone control is employed in some Motorola radio receivers. The audio systems in these receivers are designed to have a peak response at the medium frequencies, this being considered a highly desirable condition for clear voice reproduction. Means are provided for cutting down the medium frequencies and to some extent the treble frequencies when bass compensation is desired, such as at low levels of loudness. Means are also provided for cutting down the medium frequency response so as to

ser  $C_4$ . Since  $R_4$  and  $R_5$  offer no paths to ground, the entire signal voltage across  $R_2$  is applied to the control grid of the second A.F. tube through the  $R_4$ - $R_5$ - $C_5$  combination. The output of this tube feeds into the primary of an audio transformer across which is connected .007 mfd. condenser  $C_9$ . This condenser shunts some of the high-frequency audio signals around the primary, and therefore serves to cut down the high-frequency response. audio amplifier normally has reduced bass response. As a result, when the tone control is in position V, the audio amplifier system has a response which is peaked in the middle frequencies.

Since the majority of voice frequencies are in this middle region, we have a condition which many people consider ideal for the reproduction of voice.

With the tone control in position M (MUSIC), the middle frequencies are cut down considerably and bass and treble frequencies therefore predominate. Position M is therefore used chiefly for reception of music at a low level of loudness. In this position of the tone control the bass signal currents take the path through resistor  $R_4$  and condenser  $C_7$  to ground, for  $C_8$  is essentially an open circuit at low frequencies. The reactance of  $C_7$  at bass

reactance of  $C_5$  is also quite low in relation to the ohmic values of  $R_4$  and  $R_5$  so at high frequencies we can consider  $R_4$  and  $R_5$  as being in parallel. The tone control input and output voltages will thus be the same at high frequencies, but this voltage will be reduced because  $R_4$  and  $R_5$  load the detector circuit. High and low-frequency voltages are about normal, while medium frequency voltages are cut considerably. The final effect is therefore a boosting of lows and highs.

With the tone control in position B, two condensers ( $C_6$  and  $C_8$ ) are connected to ground by the switch.  $C_6$ 

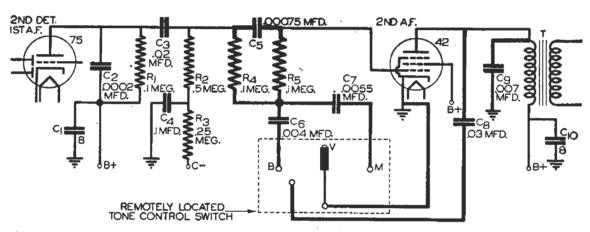


Fig. 8. Heavy lines indicate the circuit of the Acoustinator tone control system as used on the Motorola Model 70 and on many other Motorola receivers.

frequencies (about 50 cycles) is approximately .6 megohm or six times the resistance of  $R_4$ , and consequently nearly all of the bass signal voltage is developed across  $C_7$  and is impressed upon the grid of the type 42 tube through  $R_5$ . The middle frequencies, taking the same path as the bass signals, find a considerably lower reactance at the .0055 mfd. condenser; most of the available medium frequency signal voltage is wasted across  $R_4$ , and only that small portion which is developed across  $C_7$  affects the grid of the type 42 tube. At high frequencies, the reactance of  $C_7$  is so low that it can be considered a short-circuit path. The acts in much the same way as  $C_7$  in cutting down the response at medium frequencies, while  $C_8$  acts like  $C_9$  in cutting the highs. As a result, position B (BASS) gives a high bass response; this reduces static and interference noises to a minimum, and gives a soft mellowness to music.

The General Electric Tone Monitor. When an audio signal is fed from the output of an audio amplifier back into the input of an audio stage in such a way that the feed-back voltage is 180 degrees out of phase with the normal input signal for that stage, degeneration (reduction in output) will occur. In the tone control system shown in

Fig. 9, used in General Electric model F-77 and other G.E. receivers, degeneration is purposely introduced in this way to provide a control over tone.

First let us analyze circuit conditions when the tone monitor switch (a type which connects together two adjacent contacts), is in the S or speech position. Observe that the switch now shorts  $R_{16}$  and  $C_{15}$ , so that feed-back current from point 1 on the secondary winding of the output transformer flows through the switch, through resistor  $R_a$  to point 2 on the volume control, and then through section 8-G of the volume control to ground, developing across this section an out-of-phase voltage which is applied to the grid of the first A.F. stage through section z-Pof this control. The higher the feedback current, the more degeneration or reduction in signal there will be. Since there is no reactance in the feed-back current path (only the 22,000-ohm resistance of  $R_6$ ), the amount of feedback current which causes degeneration will be constant over the entire frequency range. This will lower the gain over the entire frequency range, but will not change the shape of the frequency response curve for the amplifier.

The audio frequency response curve for the amplifier circuit in Fig. 9 is that represented by curve S in Fig. 10 (the heavy solid-line curve); this will also be the response curve when the tone monitor is in the S position, for the shape of the curve is not affected by uniform reduction of gain over the entire frequency range. Note that there is a peak in the high-frequency region from 3,000 to 6,000 cycles; this occurs because the primary leakage reactance of output transformer primary  $T_5$  acts with Condenser  $C_{20}$  to form a resonant circuit at these frequencies, and maximum power is fed to the input

of  $T_s$ . The peak response in this range, along with uniform response down to 100 cycles, provides a pleasing reproduction of speech at normal room volume.

When the tone monitor switch is placed at position F, the foreign-reception position,  $R_{16}$  and  $C_{15}$  still remain shorted. Condenser  $C_{19}$  is now in parallel with resistor  $R_6$ ; at low frequencies the reactance of this condenser is quite high, with the result that feed-back current will take the path through  $R_6$ . Normal amplifier response can therefore be expected at low frequencies, for feed-back current is

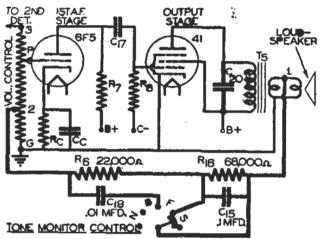


Fig. 9. The G. B. Tone Monitor Control circuit is shown here in heavy lines, as applied to the General Electric Model F-77 receiver (the amplifier circuit is shown here in simplified form).

still at the value determined by  $R_6$ . At higher frequencies the reactance of  $C_{19}$ is naturally less; since this condenser shunts  $R_6$ , more feed-back current flows at higher frequencies than would ordinarily be passed by  $R_6$  alone, increasing the degeneration and thus cutting down the high frequency response. The shape of the response curve for this foreign-reception setting is represented by curve F in Fig. 10; this differs from the normal response curve only at frequencies above about 1,000 cycles. The attenuation or cutting of the high-frequency components serves to remove objectionable noise signals which ordinarily accompany

programs received from foreign countries.

When the tone monitor switch is in the N or normal-reception position. feed-back current flows through the parallel combination of  $R_{16}$  and  $C_{18}$ , and then flows through  $R_6$  to point 2 on the volume control. At low frequencies the reactance of  $C_{15}$  is high, and  $R_{16}$  acts in series with  $R_6$  to reduce the feed-back current practically to zero. As a result, peak bass response is secured. At medium and high frequencies the reactance of  $C_{15}$  becomes so low that it shorts out  $R_{16}$ , and only Ra is effective in limiting feed-back current. This means that in the N position of the tone monitor switch we will have the normal treble peak in the response curve as well as a peak in the bass or low frequency region, as indicated by curve N in Fig. 10; this is a desirable condition for listening to music at the low loudness levels usually preferred in the home.

Setting the tone monitor switch at position B, the bass position, places condenser  $C_{19}$  across both  $R_6$  and  $R_{16}$ , and inserts the  $R_{16}$ - $C_{15}$  combination in series with  $R_6$ . At low frequencies the reactances of these two condensers are so high that we can neglect them; this leaves  $R_{16}$  and  $R_6$  acting in series and reducing feed-back current almost to zero. Degeneration is thus almost entirely eliminated at low frequencies, and bass response goes up.

At medium frequencies the reactance of  $C_{15}$  is low enough to provide a shunt path around  $R_{16}$ , but  $C_{19}$  is still out of the picture. The 22,000-ohm resistance of  $R_6$  now controls feedback current, and so we have essentially the same medium frequency response as was secured at positions S, F and N.

At high frequencies  $C_{15}$  shorts out  $R_{16}$ , and  $C_{19}$  provides a low-reactance

shunt path around  $R_6$ . Feed-back current thus increases considerably, cutting down the high-frequency response quite rapidly as frequency increases. At position B, then, curve B in Fig. 10 will be the response curve of the receiver; it clearly indicates a boosting of bass frequencies.

Naturally each setting of the tone monitor control will sound different to the human ear at each setting of the volume control, because of the changing characteristics of the human ear with loudness. Automatic compensation for the variations in sensitivity of the ear with loudness is secured by

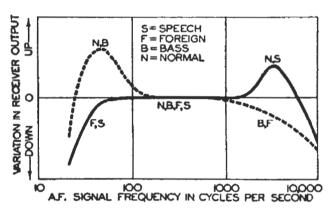


Fig. 10. This graph shows how the G. E. Tone Monitor Control (Fig. 9) changes the audio frequency response of the receiver in which it is used. The graph represents full-volume conditions, when P is at point 3; there is then no automatic tone compensation.

making the tone control connection to a tap at point 2 on the volume control potentiometer. Varying the volume control setting thus gives an automatic balance between the normal response of the amplifier and the special response introduced by the tone monitor circuit.

#### Volume Expansion Circuits

The compression of sound level is of extreme importance in the practical operation of radio transmitters, for otherwise over-modulation and an excessively low signal-to-noise ratio would exist. Compression of sound level is just as important in the pro-

duction of disc recordings of musical numbers, for excessive volume results in over-cutting of the grooves. Volume compression, of course, destroys the fidelity or faithfulness of a program, for it cuts down the volume of high-level passages. In order to counteract this volume compression, it is necessary to introduce some form of volume expansion in the reproducing device.

Although automatic volume expansion can be applied to both radio receivers and electric phonographs, it is far more satisfactory when the original compression is performed automatically, as is now being done on most recordings. In radio transmitters the compression is done manually by the studio operator, and obviously no automatic circuit in the receiver can exactly compensate for an action which depends for its accuracy upon human alertness and skill.

The most widely used system for automatic volume expansion depends upon the fact that the gain of a variable mu or super-control pentode tube varies with the negative C bias voltage which is applied. By applying to a variable mu pentode amplifier tube, in series with its normal negative C bias, a positive D.C. voltage which increases with sound level, the gain of the tube can be made to increase as sound level or volume increases. The necessary D.C. or A.V.E. (automatic volume expansion) control voltage is secured by rectifying the audio voltage and filtering out the A.C. component of the resulting pulsating voltage. This D.C. voltage must of course be time-delayed, for if the A.V.E. control voltage varies too rapidly with fluctuations in sound level, the reproduced music will sound "choppy" and have a "gurgling" effect: likewise if the A.V.E. control voltage is too greatly time-delayed, the benefit of automatic volume expansion

will be lost.

RCA A.V.E. Circuit. A typical automatic volume expansion circuit based upon this principle and used in a phonograph amplifier is shown in Fig. 11. The phonograph pick-up at the upper left in the diagram feeds an audio signal into step-up transformer  $T_1$ , across the secondary of which is shunted a 625,000-ohm volume control potentiometer  $R_4$ . Notice that this potentiometer is in an automatic tone compensating circuit, there being two taps on the potentiometer for this purpose. Series resonant circuit  $L_2$ - $C_2$ , connected to the first or uppermost tap, serves to attenuate the middle frequencies and thereby has the effect of raising the bass and treble response at medium and low volume levels. Resistor  $R_2$  broadens the resonant response of  $L_2$ - $C_2$  and limits the amount of attenuation at medium frequencies.  $R_3$  and  $C_3$ , connected between the lower tap on the potentiometer and ground, provide an additional boosting effect for the bass response at still lower volume levels. Series resonant circuit  $L_1$ - $C_1$ , also connected across the secondary of  $T_1$ , resonates at about 700 cycles and serves to cut down the response of the audio amplifier in this region. Resistor  $R_1$  prevents complete cut-off at resonance.

After passing through the volume and tone control circuits, the audio signal is fed through D.C. blocking condenser  $C_4$  into vacuum tube  $VT_1$ , which is ordinarily operated at a high negative C bias and thus normally has low gain. At the same time the audio signal developed across the secondary of transformer  $T_1$  is fed into vacuum tube  $VT_2$ , a triode amplifier; this audio signal has not been acted upon by the automatic tone and volume control circuits. The output of  $VT_2$  is rectified by  $VT_3$  and then filtered to give a D.C.

voltage whose value is dependent only upon the signal level at the secondary of  $T_1$ . This varying D.C. voltage is applied to one of the grids of pentode tube  $VT_1$  in such a way as to increase the gain of the tube in proportion to the level of the original sound. Now let us trace the circuit in greater detail.

Potentiometer  $R_8$  controls the amount of signal voltage applied to triode amplifier  $VT_2$ , and thus provides a control over the amount of volume

Since  $VT_3$  permits electron flow only in the direction from cathode to grid, the A.F. voltage existing across  $R_{10}$  sends through  $R_{11}$  a pulsating current which develops across this .22-megohm resistor a pulsating D.C. voltage, with polarity as indicated. Condenser  $C_{12}$  smooths out the A.F. variations, and the resulting D.C. voltage is applied to the third grid  $(G_3)$  of  $VT_1$  through filter  $R_{13}$ - $C_5$ , which provides the necessary time delay in the application of

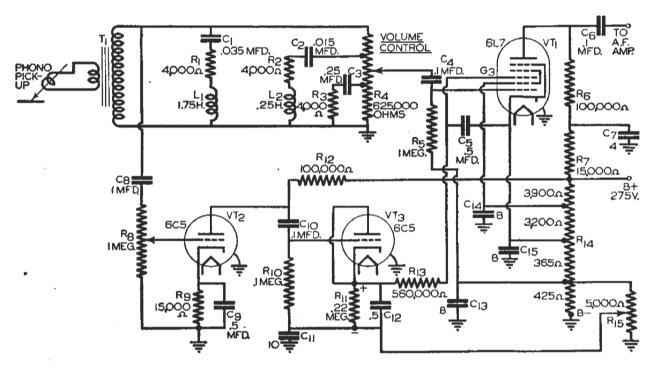


Fig. 11. Automatic volume expander circuit as used in a number of RCA electric phonographs.

expansion.  $R_9$  and  $C_9$  together provide automatic C bias for this tube, and plate voltage is obtained by a connection through resistor  $R_{12}$  to the B+terminal of the receiver power supply circuit. The A.C. plate current of  $VT_2$  flows through  $C_{10}$ ,  $R_{10}$  and  $C_{11}$  to the chassis, with most of the voltage being dropped across  $R_{10}$ . Observe that the plate and cathode of  $VT_3$  are connected together; this tube therefore acts as a diode rectifier with its grid (acting as anode) connected to one side of  $R_{10}$  and its cathode connected through  $R_{11}$  to the other side of  $R_{10}$ .

the A.V.E. control voltage. The negative terminal of  $R_{11}$  connects to ground through potentiometer  $R_{15}$ , which is connected across a part of voltage divider  $R_{14}$  in such a way as to provide an adjustable negative C bias which will act in series with the positive bias across  $R_{11}$ . Potentiometer  $R_{15}$  need be adjusted only when a new tube is inserted in the volume expander circuit.

At low volume levels practically no D.C. voltage is developed across  $R_{11}$ , and grid  $G_3$  of  $VT_1$  then receives the full negative C bias voltage developed by  $R_{15}$ . As the volume level increases,

the positive voltage drop across  $R_{11}$  increases and counteracts the negative C bias, with the result that the third grid becomes less negative and the gain of tube  $VT_1$  increases.

The action of an automatic volume expander circuit of the type shown in Fig. 11 is illustrated by the graph in Fig. 12. The lower curve tells how the output level varies with the input level when there is no automatic volume expansion, and the upper curve gives the

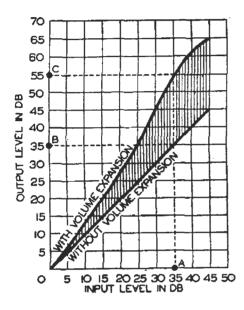


Fig. 12. Automatic volume expansion action of the phonograph amplifier circuit in Fig. 11. In preparing this graph, a definite low input signal level was assumed and called 0 db input level, and the output level under this condition was called the 0 db output level. These curves tell that if the receiver is being fed with a signal of 35 db (point A), the output level without A.V.E. will also be 35 db (point B); with A.V.E., however, the output level will be 20 db higher, or 55 db (point C).

same information for a circuit having automatic volume expansion. Notice that as the input signal level is increased, automatic volume expansion makes the output level increasingly greater than would be obtained without A.V.E. With proper circuit design, an exact reproduction of the original program can be secured even though the volume level is condensed during the recording process in the case of a phonograph record or during the transmitting and receiving process in the

case of a radio system.

Crosley Auto-Expressionator. Another interesting automatic volume expansion circuit, which in addition provides automatic bass compensation at low sound levels, is that shown in Fig. 13; as you can see, this is a special form of filter (some engineers call it a differential bridge) connected between the output transformer of the receiver and the loudspeaker. It is used in a number of Crosley receivers, where it is known as an auto-expressionator.

All four switches in Fig. 13 are operated by a single control which permits instant change-over from normal operation to A.V.E. and A.B.C. Normal operation is secured when switches  $SW_3$  and  $SW_4$  are open, and switches  $SW_1$  and  $SW_2$  are closed; a study of the circuit will show that under these conditions a direct connection exists between the loudspeaker and the output transformer, with no parts in series or shunted across the loudspeaker.

With the control switch in the A.V.E. position, switches  $SW_1$  and  $SW_2$  will be open and switches SW<sub>2</sub> and SW<sub>4</sub> will be closed, as indicated on the circuit diagram. We will assume for the moment that the loudspeaker is disconnected. Audio frequency current flowing from the output transformer to point A will have two paths to point B; 1, through  $X_1$  (a ballast resistor), resonant circuit  $L_1$ - $C_1$  and  $R_1$ ; 2, through  $R_2$ , resonant circuit  $L_2$ - $C_2$  and ballast resistor  $X_2$ . At point B the currents from the two paths combine again and return to the output transformer. Resistors  $R_1$  and  $R_2$  are equal in value; ballast resistors  $X_1$  and  $X_2$ , whose resistance increase with current, are likewise the same size; resonant circuits  $L_1$ - $C_1$  and  $L_2$ - $C_2$  are also identical in characteristics, and thus the same current would flow over each path when the loudspeaker is disconnected.

When the loudspeaker is connected between points C and D, the loudspeaker current will depend upon the voltage existing between these points. and this voltage will be the difference between the voltage drops for A-C and the voltage drop for A-D. The ballast resistors used in this circuit are of special design, increasing in ohmic value as the current through them increases. This means that as the output current of the receiver increases, the resistances of  $X_1$  and  $X_2$  increase, causing the impedance of branch A-C to be greater than that of branch A-D and causing the impedance of branch B-Dto be greater than that of B-C. Even when cold, the ohmic values of the ballast resistors are slightly greater than the ohmic values of  $R_1$  and  $R_2$ . As a result, the circuit is in an unbalanced condition at all times, with some signal voltage always being applied to the loudspeaker. The unbalance becomes greater as the receiver output current increases and sends more current through the ballast resistors. At low values of power output, when the circuit is most nearly balanced, only a small amount of the total output power is supplied to the loudspeaker.

As the power output of the receiver increases, the circuit becomes farther out of balance and more and more of the receiver output power is applied to the loudspeaker. This increases the intensity of the louder passages, expanding the volume range. The time lag in the heating and cooling of each ballast resistor filament is an important factor in the proper operation of this circuit, for it determines the speed of A.V.E. action.

Now let us see how this circuit works as a bass compensator. Since each resonant circuit resonates at about 40 cycles, we can see immediately that for the medium and high audio frequencies, the reactances of condensers  $C_1$  and  $C_2$  become quite low, and the resonant circuits thus do not create an unbalance at these higher frequencies. At low or bass frequencies, however, the resonant resistance of each resonant circuit becomes quite high, causing a greater-than-normal unbalance and thus sending more current through the loudspeaker. The result is a boosting of the bass notes at low sound levels. At high sound levels, however, the resistance of  $X_1$  increases to the point where it makes the resonant resistance of each resonant circuit prac-

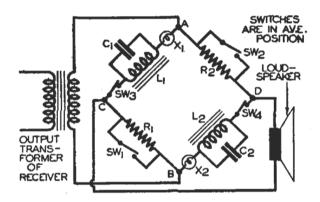


Fig. 13. Crosley Auto-Expressionator circuit, a combination automatic volume expansion and automatic bass compensation arrangement.

tically negligible in so far as unbalance is concerned; this means that bass compensation occurs only at low sound levels.

#### The Noise Problem

Signal-to-Noise Ratio. One of the most perplexing problems with which the radio engineer has to deal is that of noise. Regardless of whether this noise originates in the receiver or outside the receiver, it is reproduced along with the desired signal and is oftentimes strong enough to be annoying. Recognizing that it is impossible to eliminate noise entirely, the engineer endeavors to use apparatus which will make it negligible with respect to the desired code, sound or picture signal.

This is why he continually refers to the signal-to-noise ratio (the signal voltage or current divided by the noise signal voltage or current).

Even the average radio listener unknowingly recognizes the importance of a high signal-to-noise ratio, for he invariably tunes to a local station when he desires to enjoy a high-quality musical program with a minimum of noise. This local station, being close to the receiving location and having reasonably high power, can produce at the receiving antenna a high enough signal strength to give the desired high signal-to-noise ratio.

Man-Made Interference Noises. Noise originating outside of the receiver is usually caused by electrical apparatus such as motors, sign flashers and other devices in which sparks occur when electrical circuits are broken. This type of noise is commonly referred to as man-made interference and is most common in industrial cities and towns, where it will predominate over atmospheric noises except perhaps on stormy days. Much of this interference can be reduced by the use of special noise-reducing antennas (considered elsewhere in the Course), but the ideal remedy is elimination of the interference at its source by the use of noise filters.

Radio stations have aided materially in reducing the effects of noise by increasing the power of their transmitters. In extremely noisy locations in cities, owners of radio receivers should be told to tune to the high-power local stations in order to secure a high signal-to-noise ratio at their receivers.

The wave form of a typical noise signal is shown in Fig. 14; notice that there is a general noise level, existing practically all of the time, which is ordinarily not objectionable except when low-power or distant stations are

tuned in; in addition, there are sharp pulses of noise or static at irregular intervals, which usually are objectionable. As a rule these pulses are of very short duration, less than .001 second.

Receiver Noise. Inside the radio receiver, thermal agitation and tube shot effect, particularly in the first R.F. stages of a receiver, result in unavoidable circuit noises. Fortunately these circuit noises can be reduced to a level less than that produced by a one-microvolt input signal (by proper selection of circuits and tubes), and the signals of most stations can override this noise level. The mixer-first detector in a superheterodyne receiver can give excessive noise, but a properly

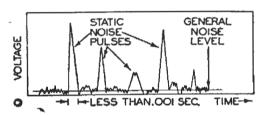


Fig. 14. Wave forms of typical rectified noise signals as they may exist in the detector stage of a radio receiver.

operated stage preceded by R.F. amplification will not have this trouble.

We shall now consider in turn the use of tone control circuits, noise impulse-silencing circuits and inter-carrier noise suppression circuits for noise-reducing purposes.

#### Noise-Reducing Tone Controls

An analysis of noise signals will reveal that they contain practically all audio signal frequencies, with the higher frequencies, above 3,000 cycles, predominating. It is for this reason that a tone control which suppresses or attenuates the high audio frequencies is also effective in reducing the amount of noise. Of course, this suppression of the high audio frequencies destroys the faithfulness of reproduction, and consequently tone controls

are far from being a satisfactory solution to the noise problem.

#### Noise Impulse-Silencing Circuits

The Lamb Noise Silencer. The circuit shown in Fig. 15, developed by J. J. Lamb, serves to eliminate noise pulses by silencing the receiver for the duration of each pulse. This circuit has

generally inserted between the I.F. amplifier and the second detector of a superheterodyne receiver, as indicated in Fig. 15.

The signal voltage developed by the second I.F. transformer between point A and ground is made up of an I.F. carrier modulated by noise signals and by desired intelligence signals. This voltage is applied directly to the con-

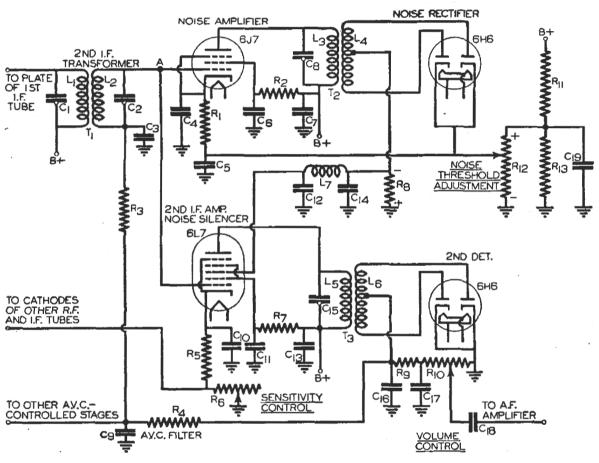


Fig. 15. The Lamb noise silencer, which silences the receiver for the duration of each strong noise pulse, is shown here incorporated in a superheterodyne receiver.

been found quite valuable in commercial and communication receivers which are used for code signal reception. It has not been entirely satisfactory for broadcast receivers, however, for the action of the noise-silencing circuit has the effect of partially destroying the peaks of modulation in the received carrier, thereby creating the more serious problem of amplitude distortion.

The Lamb noise-silencing circuit is

trol grid of the 6J7 noise amplifier tube, and also to the control grid of the 6L7 second I.F. amplifier tube. The amplified output of the second I.F. tube feeds through tuned primary transformer  $T_3$  to the 6H6 full-wave diode second detector, which through rectification provides the desired A.F. signal voltage and a D.C. voltage for A.V.C. purposes. The amplified output of the noise amplifier tube, on the other hand, feeds through tuned primary

transformer Tax (tuned to the I.F. value) to another 6H6 double-diode tube which serves as a full-wave noise rectifier. The cathodes of both the noise amplifier and noise rectifier tubes are connected to the movable contact of noise threshold adjustment potentiometer  $R_{12}$ , and the setting of this control determines the value of the negative bias which is applied to the control grid of the noise amplifier and to the diode sections of the noise rectifier. This bias on the diodes acts much like a delay voltage, in that it determines the minimum signal strength required to swing the diode plates positive and start rectification; the bias is normally set so that rectification occurs only for noise peaks extending above the desired signal level. Note that the noise amplifier tube also receives a bias from the A.V.C. circuit of the receiver through A.V.C. filter  $R_3$ - $C_3$ ; this serves to maintain the proper noise threshold level despite normal variations in desired signal level due to fading.

The rectified noise voltage, which is developed across noise rectifier load resistor  $R_8$  when strong noise peaks are present, is made up of I.F., A.F. and D.C. components. Filter combination  $C_{14}$ - $L_7$ - $C_{12}$  removes the I.F. and A.F. components; the remaining D.C. voltage is applied to the third grid of the second I.F. amplifier-noise silencer tube, driving this grid more negative with respect to ground and thus blocking plate current flow for the duration of each strong noise pulse.

When no noise pulses are present, the noise rectifier plates are negative with respect to their cathodes; no current passes, no D.C. voltage exists across  $R_8$ , and consequently the third grid of the 6L7 second I.F. tube has only the normal negative C bias determined by  $R_5$  and  $R_6$ . Desired signals are therefore amplified and de-

tected in a normal manner by the two lower tubes in Fig. 15.

When a strong noise pulse enters the receiver along with the desired signals, it instantly drives the noise rectifier plates positive, and the D.C. voltage developed across  $R_8$  by the resulting rectified current flow is applied without delay to the third grid of the second I.F. amplifier-noise silencer tube, blocking plate current flow completely. Under this condition no desired or noise signals whatsoever reach the second detector, and consequently the noise is not heard. The instant the noise pulse drops below normal signal level, the noise rectifier stops conducting, the blocking voltage is removed from the third grid of the 6L7 tube, and normal receiver operation is restored.

Naturally there must be no time delay between the arrival of a noise pulse at the noise amplifier and the application of the D.C. silencing voltage to the third grid of the noise-silencer tube if the silencing action is to be instantaneous and is to last only for the duration of each strong noise pulse. Under this condition the receiver is silenced for such a short interval at a time that the action can scarcely be noticed by the human ear.

Diode Noise Limiter. If the load of a diode detector is instantly shunted with a large condenser when a noise pulse comes through, the audio voltage developed across this load, as well as the noise pulse, will be greatly reduced for the duration of the pulse. A practical noise-limiting circuit based upon this principle is shown in Fig. This circuit uses a type 6H6 double-diode tube, with section  $D_1$ serving as second detector (resistors  $R_2$  and  $R_3$  form its load) and section  $D_2$  serving as an automatic noiselimiter switch places a large condenser

across  $R_3$ , a part of the detector load, during the time of a noise pulse (the two sections are shown as separate tubes to simplify your study of the circuit, but remember that they are actually in the same vacuum tube envelope). The D.C. voltage developed across the detector load resistors is used for A.V.C. purposes, with  $C_F$  and  $R_{\rm F}$  serving as the A.V.C. filter. Condenser C is an R.F. by-pass condenser which keeps I.F. signals out of the detector load. Switch SW is a panel control, and must be in the closed position when noise-limiting operation is desired.

When no strong noise pulses are present, the normal rectified electron flow is from point 1 to point 3 through the detector load, and point 1 is therefore more negative than points 2 and 5. The D.C. voltage developed between points 1 and 3 charges condenser  $C_1$  through resistor  $R_1$ , which is of high ohmic value. Once this condenser is charged. no additional current flows through  $R_1$ , and points 1 and 4 are then at the same negative potential with respect to ground (point 3). Diode section  $D_2$  now acts as a high resistance, for its anode (point 4) is more negative than its cathode (point 2).

Now suppose that a strong noise pulse is rectified by diode section  $D_1$ along with the desired modulated car-The flow of rectified current through the detector load increases instantly, and the D.C. voltage between points 1 and 3 will likewise increase instantly.  $C_1$  will begin drawing current through  $R_1$  in order to become charged to the new voltage value. The time constant of the  $R_1$ - $C_1$  combination is quite high, however (ordinarily it is from 1/10 to 1 second), and consequently  $C_1$  cannot be charged instantly to the new voltage. Since the average noise pulse lasts about 1/1000

of a second, which is considerably less than the time constant, point 4 remains essentially at the previous low negative potential for the duration of the noise pulse, while point 2 goes instantly to the new higher negative potential. This means that the plate of diode section  $D_2$  becomes positive with respect to its cathode for the duration of the noise pulse; current then flows through  $D_2$  and its resistance drops to a quite low value. Since this tube is in series with high-capacity condenser  $C_1$  across  $R_3$ , the net impedance between points 2

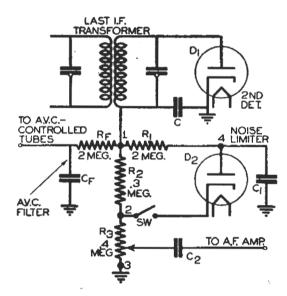


Fig. 16. Diode noise-limiter circuit, which shunts all audio signals to ground through a condenser for the duration of each strong noise pulse when switch SW is in the closed or noise-reducing position.

and 3 is lowered, and the A.F. voltage across  $R_3$  is therefore lowered. As a result, the A.F. voltage fed to the first audio stage drops considerably for the duration of each noise pulse, and the A.F. output voltage of the receiver is correspondingly lowered. The instant the noise drops below normal signal level again, point 2 returns to its original potential, current flow through  $D_2$  stops, and the  $D_2$ - $C_1$  shunt path across  $R_3$  is no longer effective in limiting the audio amplifier input voltage.

#### Inter-Carrier Noise Suppression Circuits

In receivers having A.V.C., the noise which is heard when tuning between stations may be objectionable to some listeners. This inter-station or intercarrier noise occurs because automatic volume control raises the gain of the R.F. system to the point where normal) external and internal noises become objectionable. Some receivers have an inter-station noise-suppression circuit which works in conjunction A.V.C. to silence the receiver when no station is tuned in. This inter-carrier noise suppression system is often known as quiet automatic volume control, abbreviated Q.A.V.C.; since it stops or squelches noise it is also known as a noise-squelching circuit. In some circuits the detector action is delayed for weak signals; in other circuits the. I.F. or A.F. amplifier is blocked until the station signal reaches a level which is high enough to override noise which may be present, thus preventing reception of stations having less than the minimum satisfactory signal-to-noise ratio. To be sure, this has the effect of reducing the apparent sensitivity of the receiver; a switch is therefore provided to cut out this noise-suppressing action when maximum sensitivity is required regardless of noise.

diode noise-limiter circuit The shown in Fig. 16 also serves to a certain extent as an inter-carrier noise suppressor. Suppose that the receiver is tuned off a station. Diode section  $D_1$ of course receives no input voltage under this condition, and no current flows through  $R_2$  and  $R_3$ . Points 1 and 2 immediately assume ground potential, and point 4 will likewise be at ground potential after the time-delay interval (less than one second), during which condenser  $C_1$  discharges to ground through  $R_1$ ,  $R_2$  and  $R_3$ , has elapsed.

Noise or static signals which enter the receiver under this condition will, of course, be rectified by diode section  $D_1$ , making point 2 negative with respect to ground. Point 4 remains at ground potential because of time-delay action, making the plate of  $D_2$  positive with respect to its cathode. Noise signals reaching point 1 will therefore take the low-impedance path through  $R_2$ ,  $D_2$ and  $C_1$  to ground. Noise is thus diverted from the input of the A.F. amplifier and will not be reproduced by the loudspeaker when tuning slowly from station to station. Of course, the noise-reducing action of this circuit ceases as soon as the time-delay interval has elapsed.

In the more conventional types of inter-carrier noise suppression circuits, the receiver is made inoperative for weak signals on the assumption that the noise signals heard between stations will be weaker than any of the desired strong carrier signals. types of inter-carrier noise suppression circuits are used for this purpose: 1, biased demodulator circuits, which place a bias or delay voltage on the diode detector so that only carrier signals above a certain level will be demodulated; 2, A.F. amplifier blocking circuits, which block or prevent operation of the A.F. amplifier until the carrier signal exceeds a definite minimum value; 3, I.F. and R.F. amplifier blocking circuits, which prevent operation of these amplifiers until the carrier level becomes appreciably higher than the noise level. Let us consider one example of each of these inter-carrier noise suppression methods.

Biased Demodulator Circuit. As you know, delayed A.V.C. action can be obtained by inserting in the separate A.V.C. diode rectifier circuit a delay voltage which must be exceeded by the carrier signal before the rectification

required for the production of an A.V.C. voltage can take place. Usually, a separate diode rectifier is then required for demodulation purposes.

If we introduce this delay voltage directly into the diode rectifier circuit used for demodulation, so that only signals above a certain minimum level will be rectified, we have one form of inter-carrier noise suppression. Weak carrier and noise signals cannot overcome this delay voltage, and hence are not reproduced by the loudspeaker as the receiver is tuned from one station With this arrangement, to another. however, the receiver must be tuned to fairly strong carriers in order to secure satisfactory reproduction, for mediumstrength signals will undergo a certain amount of amplitude distortion.

A circuit incorporating this biased demodulator type of inter-carrier noise suppression is shown in Fig. 17. In the D (distance) position of switch SW, the two diode sections of the 6B7 tube together serve for demodulation and A.V.C. purposes, and there is no delay voltage in the diode circuit. Under this condition all signals, including noise, are detected in the usual manner.

When the switch is placed in the L position, the voltage drop across cathode bias resistor  $R_{\rm c}$  acts in series with the diode circuit. This delay voltage must be exceeded by the signal before A.V.C. action and demodulation can take place, and the carrier signal must be reasonably larger than this delay voltage before complete demodulation can occur. If  $R_{\rm c}$  (which acts also as C bias resistor for the amplifier section of the 6B7 tube) is replaced with a potentiometer and the movable arm is connected to the L terminal of the switch, the amount of delay voltage in the rectifier circuit can be varied.

A.F. Amplifier Blocking Circuit. In

Fig. 18 is a well-known example of a circuit which employs quiet automatic volume control (Q.A.V.C.) action to block or cut off the input signal to the A.F. amplifier whenever the carrier level becomes so low that noise will be objectionable. As you can see, a special Q.A.V.C. tube is inserted between the detector and the first A.F. stage, with a switch in the cathode lead to make this tube inoperative when ordinary receiver action is desired.

Observe that the plate-cathode resistance of the Q.A.V.C. tube acts in series with a 1-megohm resistor (part

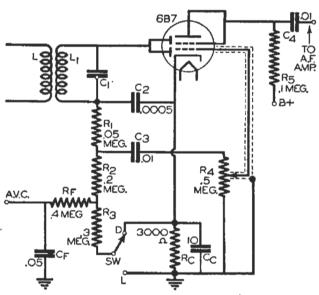


Fig. 17. Biased demodulator type of inter-carrier noise suppression circuit, as used in the Silvertone Model 7149 receiver. This circuit reduces the noise ordinarily heard in an A.V.C.-controlled receiver when tuning between strong stations.

72) between B+ and ground G, with the screen grid of the type 77 A.F. amplifier tube connected to point X as indicated. We thus have a voltage divider, with the screen grid voltage of the type 77 tube being determined by the plate-to-cathode resistance of the Q.A.V.C. tube. The lower this resistance, the lower will be the positive voltage developed across the Q.A.V.C. tube and applied to the screen grid of the A.F. tube, and the lower will be the gain of this type 77 tube.

Now let us see how weak signals lower this plate-cathode resistance and

thereby block the first A.F. tube so as to suppress inter-carrier noises. Notice that the control grid of the Q.A.V.C. tube connects to the diode second detector load through a 4-megohm resistor. Under a no-carrier-signal condition, the only D.C. voltage across diode load resistor 55 will be a very low D.C. voltage produced by noise signals; condenser 52 and resistor 53 remove all A.C. component which might otherwise reach the Q.A.V.C. tube. The Q.A.V.C. tube thus gets a small negative C bias, under which condition its plate current is high and its plate-to-cathode resistance quite low. This makes the screen

this tube and thereby increasing the screen grid voltage on the 77 tube enough to permit amplification of the desired signals in the usual manner.

In the circuit just described, blocking action was secured by varying the screen grid voltage of the first A.F. amplifier tube; the same results can be secured by applying the blocking voltage to the control grid of this tube, as is done in the inter-carrier noise suppression circuit in Fig. 19. No extra tube is required here, for the triode section of double-diode-triode tube  $VT_1$  is used to produce the required blocking voltage. A desired carrier signal

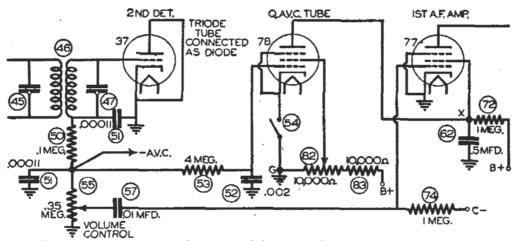


Fig. 18. This quiet A.V.C. circuit, used in the Philos Model 16-122 receiver, acts upon the acreen grid of the first A.P. tube, blocking the amplifier whenever the receiver is tuned between strong stations. It thus serves to suppress inter-carrier noise.

grid voltage of the type 77 tube low enough to prevent amplification of any noise signals which are fed to the control grid of this tube through D.C. blocking condenser 57. Potentiometer 82 controls the screen grid voltage on the Q.A.V.C. tube and thus determines the amount of plate current which will flow during this low negative C bias condition. This potentiometer can be adjusted so that signals below any desired level will be blocked.

When a desired strong carrier signal reaches the detector, the resulting D.C. voltage across detector load resistor 55 increases the negative bias on the control grid of the Q.A.V.C. tube, increasing the plate-to-cathode resistance of

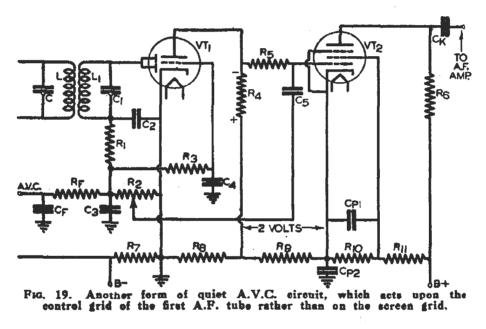
is rectified by the combined diode sections of the tube in the usual manner, developing a D.C. voltage across detector load resistor  $R_2$ . The audio voltage which also exists across this resistor is fed through D.C. blocking condenser  $C_5$  to the control grid of the pentode first A.F. amplifier tube VT<sub>2</sub>, while the D.C. voltage is fed through resistor  $R_3$  to the control grid of the triode section of tube  $VT_1$ . Plate current for this triode section flows through resistor  $R_4$ , producing across it a voltage drop which acts with the drop across voltage divider resistor  $R_2$ to provide C bias for  $VT_2$ . You can readily see that when the current through  $R_4$  is small, the C bias on the

pentode tube will essentially equal the voltage drop across  $R_0$  which is ordinarily about 2 volts. This is the C bias required for normal A.F. amplification of a desired signal.

When no carrier signal is being rectified by the diode section, the D.C. voltage across  $R_2$  is essentially zero, which means that the triode section of  $VT_1$  will be operating at zero bias. This makes the plate current of this section high, and this large current flowing through  $R_4$  increases the negative C bias considerably on the control grid of  $VT_2$ . As a result, the pentode can-

A.F. amplifier blocking circuit. In Fig. 20 is shown a circuit which, when switch SW is closed, will accomplish the desired results, driving all of the control grids in the R.F. and I.F. sections highly negative for weak signals and placing a negative bias on the second detector diode sections to prevent rectification of weak signals. Normal bias is restored immediately when the carrier signal becomes greater than the minimum desired level.

An analysis of this circuit starts with resistor  $R_{13}$ , across which the second detector develops the desired audio



not amplify any weak signals or noise signals which may appear across  $R_2$  as the receiver is tuned between stations.

When a desired strong carrier signal is tuned in, the D.C. voltage developed across  $R_2$  drives the grid of the triode highly negative, reducing the plate current of  $VT_1$  and restoring the C bias of  $VT_2$  to its normal value.

I.F. and R.F. Amplifier Blocking Circuit. Noise signals and weak carrier signals which are ordinarily heard when tuning between stations can be suppressed by applying to the I.F. and R.F. sections the same principle as that just described in connection with the

voltage and the D.C. voltage for A.V.C. purposes; the plate current of the whisper control tube also flows through  $R_{13}$ , developing across it an additional voltage for blocking purposes. When no carrier signal is present, the only voltage across  $R_{13}$  is that due to plate current of the whisper control tube, and the magnitude of this voltage is determined by the setting of movable arm P on potentiometer  $R_{13}$  and by the negative bias on the control grid of this tube. Point 1 will be negative with respect to the chassis during this nocarrier-signal condition.

Now observe that the output of the I.F. amplifier is fed to the upper diode

of the whisper control tube through  $C_{19}$ . When no carrier signal is present, there will be no rectified current flow, and point X and the control grid of the whisper control tube will therefore be at cathode potential; as a result a large plate current flows through the triode section of this tube, then through SW,  $R_{13}$ ,  $R_{16}$ ,  $R_{17}$ ,  $R_{18}$  up to P and back to the cathode. This develops a large negative voltage across  $R_{13}$  and since this voltage is applied to the A.V.C.-controlled tubes through the usual A.V.C. feed system, it blocks these tubes.

When weak intelligence and noise signals are present in the output of the I.F. amplifier, such as when tuning between powerful local stations, these pass through  $C_{19}$  to the upper diode section of the whisper control tube. where they undergo rectification and produce D.C. voltage drops across resistors  $R_4$ ,  $R_{29}$  and  $R_7$ . The direction of electron flow is such that point X on  $R_7$  will be negative with respect to the cathode of the tube; since the control grid of the triode section of the whisper control tube is also connected to this point, it gets a negative C bias. Plate current flow is thereby reduced. and the voltage drop produced by this plate current is correspondingly reduced. This reduces the negative bias applied to the R.F. and I.F. sections through the A.V.C. system, but the negative bias on the second detector diode sections is still high enough to prevent signal rectification.

When a sufficiently strong carrier signal is tuned in, the diode sections of the whisper control tube will pass enough current to block plate current flow through the triode section (by increasing the voltage drop across  $R_7$ ), and practically no current will be drawn through  $R_{13}$  by the whisper control tube. This com-

pletely releases the delay action of the whisper control tube upon the A.V.C. circuit, restoring normal high sensitivity of the receiver. A.V.C. action remains normal with further increases in earrier level. The setting of potentiometer  $R_{18}$  determines the minimum carrier level at which the delay action is released and normal A.V.C. action begins.

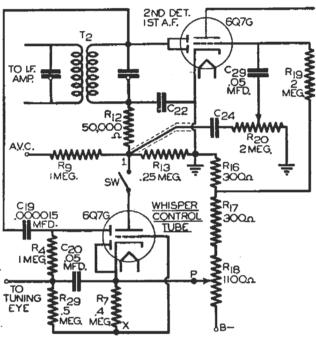


FIG. 20. This inter-carrier noise suppression circuit used in the Silvertone Model 4666 receiver where i is called a "whisper tuning control," serves to block the R.F. and I.F. amplifiers through the A.V.C. system, and at the same time provides delayed A.V.C. action.

The circuit in Fig. 20 thus provides a combination of inter-carrier noise suppression and delayed A.V.C. action. The listener is able to tune from one station to another even at low sound levels or whisper levels without hearing inter-station noise, and thus this circuit has been called a whisper control circuit. Opening switch SW removes the delay action, allowing reception of weak signals along with noise. Observe that the opening of this switch does not stop the functioning of the diode section of the whisper control tube; this is essential because the diode also provides a D.C. control grid voltage for the electron ray indicator tube or tuning eye.

## Lesson Questions

Be sure to number your Answer Sheet 24FR-1.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

- 1. What three types of circuits are used for reducing the annoying effects of atmospheric noises which enter a radio receiver?
- 2. At low loudness levels, is the human ear more sensitive to medium-frequency sounds (500 cycles to 5,000 cycles) than to low-frequency sounds (below 500 cycles)?
- 3. Why does a simple condenser tone control give an apparent boost in bass signals?
- 4. How can a conventional tone control provide treble control?
- 5. What type of circuit is present when the volume control has a fixed tap which is connected to ground through a condenser and a resistor?
- 6. Why is degeneration intentionally introduced in the General Electric Model F-77 receiver circuit shown in Fig. 9.
- 7. Upon what fact does the operation of the most widely used automatic volume expansion system depend?
- 8. Why should a receiver be tuned to a local station when a minimum of noise is desired?
- 9. Why is inter-carrier noise heard in receivers having A.V.C.?
- 10. Name the three conventional types of inter-carrier noise suppression circuits.